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SPECIFICATION

METHOD AND APPARATUS FOR MEASURING THE TRANSMISSION QUALITY OF A TRANSMISSION CHANNEL

The invention is directed to a method and to a corresponding apparatus for measuring the transmission quality in a transmission of digital information via a transmission channel.

The need for digital transmission systems has exponentially risen in recent decades. Digital transmission systems are generally classified into the function units shown in Fig. 1. A message source 1 generates information that is transmitted by a transmitter via a transmission channel 4 to a receiver. The properties of the information to be transmitted are dependent on the message source. Messages to be transmitted can, for example, be an audio signal or a video signal. Analog transmission signals thereby transmit analog signals that were generated by analog message sources, transmitting these directly via the transmission channel upon employment of traditional analog modulation methods. Such modulation methods are, for example, amplitude modulation, frequency modulation or phase modulation. In digital transmission systems, the information to be transmitted is converted into a sequence of binary numbers. In order to be able to utilize the capacity of the channel optimally well, the message to be transmitted should be represented with as few binary numbers as necessary. To this end, a source encoder is employed that has the job of converting the messages to be transmitted into sequences of signal values and encoding them, so that the channel can transmit them. The source encoder thereby attempts to convert the messages to be transmitted into binary numerals as efficiently as possible. The sequence of binary numbers generated by the source encoder is transmitted by the channel to the receiver. Such an actual channel can, for example, be composed of a line connection, of a coaxial cable, of a light waveguide (LWL), of a radio connection, a satellite channel or a combination of these transmission media. Such channels cannot directly transmit the sequence of binary numbers from the transmitter. To that end, the sequence of digital information must be converted into signal values that correspond to the properties of the channel. Such a device

is called a digital modulator. Such a modulator is part of the channel encoder 3, which additionally comprises a discrete channel encoder in order to provide the information to be transmitted with an error protection adapted to the channel.

It is not assumed of the transmission channel 4 that it works error-free; rather, it is assumed that a noise source 5 will modify the transmitted signals during the transmission with a specific probability.

Such disturbances can, for example, be a cross-talk of signals that are transmitted on neighboring channels. The disturbances can likewise be caused by thermal noise that is generated in the electronic circuit such as, for example, amplifiers and filters that are employed in the transmitter and in the receiver. Given line connections, disturbances can additionally be caused by switchings and can be additionally caused by meteorological influences given radio or satellite connections such as, for example, thunderstorms, hail or snow. Such influences modify the transmitted signal and cause errors in the received digital signal sequence.

In order to nonetheless assure a relatively dependable transmission, the channel encoder increases the redundancy of the (binary) sequence to be transmitted. With the assistance of this redundancy added by the transmitter, the receiver is assisted in the decoding of the information-carrying signal sequence. To this end, for example, the channel encoder combines a specific plurality of signals to form blocks and a plurality of check signals (one parity bit in the simplest case) is added. In this way, k information bits are always simultaneously encoded, whereby each k bit sequence has an unambiguous n bit sequence, what is referred to as the code word, allocated to it. The redundancy added in this way can be indicated with the ratio n/k. This likewise corresponds to the channel bandwidth that must be correspondingly increased in order to transmit the information sequence expanded by the added redundancy.

Alternatively, an enhanced dependability against channel disturbances can also be achieved, for example, by an increase in the transmission power. Since the increase in the transmission power, however, is relatively expensive, the dependability is usually achieved given available bandwidth by increasing the required channel bandwidth.

In the transmission of one bit with the data rate R bit/s, the modulator always allocates a signal curve or, respectively, a signal value (referred to below only as signal value) $s_1(t)$ to the binary number 0 and allocates a signal value $s_2(t)$ to the binary number 1. This transmission of each individual bit by the channel encoder is called binary modulation. Alternatively, the modulator can simultaneously transmit k information bits upon employment of $M = 2^k$ different signal values $s_i(t)$ with i = 1, 2, ... M, whereby each of the 2^k possible k-bit sequences is allocated to a signal value.

At the receiver side of a digital transmission system, the digital demodulator processes the signal value transmitted in the channel (potentially modified) and allocates an individual number to each signal value that represents an estimate of the transmitted data symbol (for example, binary).

After reception of a signal in the receiver, the demodulator must decide which of the M possible signal values was sent. This decision is implemented in a decision unit (slicer), whereby the decision should be made with minimal error probability. This decision unit allocates a reception value (usually edited) to one of the M possible symbol values.

When, for example, a binary modulation is employed, the demodulator must decide when processing each received signal whether the transmitted bit is a matter of a 0 or of a 1. In this case, the demodulator implements a binary decision. Alternatively, the demodulator can also implement a ternary decision, whereby the demodulator decides for "0", "1" or "no decision" dependent on the quality of the received signal.

The decision process of a demodulator can be viewed as quantization, whereby binary and ternary decisions are specific instances of a demodulation that quantizes Q-level, whereby $Q \geq 2$ applies. In general, digital communication systems employ a high-order modulation, whereby m = 0, 1 ... M-1 represents the possible transmitted symbols.

When the transmitted information contains no redundancy, the demodulator must decide at every predetermined time interval which of the M-signal values was transmitted. When the transmitted information, in contrast, contains redundancy, then the demodulator reconstructs the original information

sequence on the basis of the code employed by the channel encoder and on the basis of the redundancy of the received signals. Dependent on the demands defined by the applications, the channel encoder generates signal blocks that make it possible for the channel decoder to either only identify where the specific disturbances have occurred (error-recognizing encoding) or to even be able to automatically correct (error-correcting encoding) errors caused by disturbances (up to a specific maximum number per signal block).

One criterion for the dependability with which the messages are transmitted from the transmitter to the receiver is represented by the error rate. The error rate indicates the average probability with which a bit error occurs at the output of the decoder. The bit error rate indicates the plurality of error bits occurring at the receiver divided by the total number of received bits per time unit. The bit error rate (or symbol error rate when the error frequency of symbols is evaluated) is the most important quality criterion of a digital transmission system. In general, the error probability is dependent on the code properties, on the nature of the signal values employed for the transmission of the information via the channel, on the transmission power, on the properties of the channel, i.e. the strength of the noise, the type of noise, etc., and on the demodulation and decoding method. The significance of the bit error rate for digital transmission systems corresponds to the signal-to-noise ration (SNR) of analog transmission systems.

Traditionally, a known bit sequence or, respectively, symbol sequence is transmitted at periodic time intervals for determining the error rate, being transmitted in addition to the transmitted information and also be known to the receiver. Such a signal is generally composed of a pseudo-random sequence of suitable length. The error rate can be determined in the receiver in that a comparison of the transmitted signal to the received signal is implemented (rated-actual comparison).

An object of the invention is to create an improved method and an improved apparatus for measuring the transmission quality of a digital transmission channel.

This object is achieved for a method with the technical teaching of claim 1 and is achieved for an apparatus with the technical teaching of claim 7.

Advantageous developments of the invention are recited in the subclaims.

Inventively, a signal value is again allocated to each detected symbol in the demodulator at the receiver side, namely a signal value that the input of the decision unit in the demodulator would have received if the signal curve or, respectively, signal value corresponding to the detected signal had been transmitted unfalsified. In this way, a hypothetical input signal corresponding to the detected symbol values is formed that contains no signal values with channel distortions. This reference signal, as long as the decision unit does not detect false symbols, thus corresponds to the original signal at the transmission side. By subtracting this reference signal from the actually received signal, the noise signal can be acquired. With the assistance of these two signal parts, the quality of the transmission channel can be defined. The average power of this reference signal formed in this way thus corresponds to the average power of the received, undisturbed signal part. The average power of the received signal corresponds to the combination of disturbed and undisturbed signal parts. With the assistance of the previously calculated, undisturbed signal part, the reference signal, the noise power is calculated therefrom. The signal-to-noise ratio (SNR) derives from the ratio of the average power of the undisturbed signal part to the average power of the noise part, deriving as a criterion for the transmission quality of the transmission channel.

What this invention particularly avoids is that the receiver must know a specific transmission sequence, as necessary in traditional methods. Moreover, the determination of the error rate ensues parallel with the evaluation of the transmitted symbols, i.e. online. A periodic introduction of a measuring sequence into the data stream to be transmitted is therefore no longer required for the continuous measurement of the transmission quality. In this way, a reduction of the net data rate of the transmission channel can be avoided.

In order to assure a high statistical dependability, the traditional method that employs a test sequence known to the transmitter and receiver must acquire a great plurality of errors, usually a few hundred. For the extremely low bit error

rates of, for example, 10^{-9} that are generally required, the traditional methods need very long measuring times in order to detect a corresponding plurality of errors. The inventive method, in contrast, is based on the interpretation of the measured signal-to-noise ratio during the ongoing transmission. Since, however, significantly shorter measuring times are required for the interpretation of the average powers than for the comparable evaluation of the symbol stream or, respectively, bit stream of the test sequences, the transmission quality can be identified far, far faster with the inventive method.

The invention thus enables a monitoring of the actual error rate at noticeably shorter time intervals since the actually transmitted information cannot be employed traditionally for determining the error rate and, thus, one must wait for the occurrence of transmission errors in the test sequences that are only rarely introduced.

In a further development of the invention, the identified transmission quality, the signal-to-noise ratio (SNR), can be converted into a symbol or, respectively, bit error rate dependent on the respectively employed encoding method.

Preferred exemplary embodiments of the invention are explained next with reference to the drawing. Shown are:

Fig. 1 the general structure of a message transmission system;

Fig. 2 the structure of an inventive receiver;

Fig. 3 the structure of an inventive demodulator of the receiver shown in Fig. 2;

Fig. 4 the structure of devices for determining the transmission quality of the transmission channel in the receiver shown in Fig.2;

Fig. 5 a device for allocating an identified transmission quality to an error rate in the receiver shown in Fig. 2; and

Fig. 6 a diagram of characteristics for the allocation of a signal-to-noise ratio to the probability of a symbol error dependent on the modulation method employed.

In digital information transmission, information are transmitted between a message source (transmitter) and a receiver via a transmission medium. Such an

apparatus that is located between the transmitter and the receiver is generally referred to as channel.

For the transmission, the data to be transmitted are converted into code words that are matched to the transmission properties of the message channel in order to protect the data to be transmitted against, among other things, transmission errors.

In the transmission, a character, which is generally referred to as symbol in the signal space or channel symbol, is allocated to a bit sequence with a reversibly unambiguous, functional allocation. This symbol is subsequently mapped onto a signal curve (referred to below as signal value) allocated to this symbol. The functional allocation of a symbol to a bit sequence in the transmitter is called encoding or mapping; the mapping of such a symbol or of a plurality of such symbols onto a signal value is called modulation.

The reversal of this mapping sequence occurs in the receiver. During the demodulation, i.e. the allocation of a reception signal to a symbol, can usually not be implemented error-free due to distortions or superimposed disturbances of the channel, the decoding, i.e. the conversion of a detected symbol into the corresponding bit sequence does not represent any problems. Fig. 2 shows an inventive receiver that comprises a demodulator 10, a signal-to-noise ratio identification means 11 and an error rate identification means 12. The demodulator processes the received signal 13 in order to output a corresponding bit sequence 16 at its output. Such a demodulator 10 contains a decision unit 18 that allocates one or more symbols 9 or, respectively, the corresponding signal value 15 to the edited reception value 14 following the analog and the optional first steps of the digital signal processing (combined here to form the block "signal editing" 17). The signal-to-noise ratio identification means 11 shown in Fig. 2 contains two different identification devices 20, 21 in order to identify a signal-to-noise ratio 22. An error rate 23 is allocated to the identified signal-to-noise ratio 22 in the error rate identification means 12 dependent on the respective encoding method.

Fig. 3 shows the structure of an inventive demodulator in the receiver of a digital transmission system. The signal 13 received from the transmission

channel 4 is supplied to a signal editing device 17 that, for example, contains the analog-to-digital conversion needed for the digital signal processing and/or a distortion correction of the transmitted signals. The edited signal values 14 are subsequently supplied to the position unit 18 that, using this signal value, decides which symbol or symbols were most probably transmitted. The selected symbol or symbols 9 are conducted to the decoder 19 by the decision unit, said decoder 19 converting the symbols 9 into the bit sequence 16.

The representation of the symbol values at the output 15 of the decision unit 18 shown in Fig. 2 or, respectively, Fig. 3 is identical to the corresponding signal values of the detected symbol, i.e. the signal values predetermined by the modulation in the transmitter. This signal value sequence 15 which is based on the detected symbols 9 is simultaneously forwarded - together with the detected signal value 14 - to a signal-to-noise ration identification means 11 and/or to the preceding signal editing unit 17.

Such a signal-to-noise ratio identification means 11 is shown in Fig. 4. The illustrated identification means contains two versions (version 1, version 2) for calculating the signal-to-noise ratio 22. In an inventive receiver, it suffices to identify the signal-to-noise ratio in only one way.

Whereas the detected signal values 14 contain a signal part and a noise part, the signal values 15, which were identified based on the detected symbols 9, contain only the signal part. In both alternatives (version 1, version 2), the signal part S is divided by the noise part N (noise) in a division device 28 in the signal-to-noise ration identification means 11. To this end, the average signal part S and the average noise power N must be respectively present independently of one another. The average signal power S is identified in the device 24 for determining the average power, being identified from the signal values 15 both according to version 1 as well as according to version 2.

For determining the noise power N, the signal part must be subtracted from the combined signal and noise part of the signal values 14. To that end, the signal values of the reference signal 15 are subtracted from the detected signal values 14 in the first embodiment (version 1) in order to obtain the noise signal

values. The noise signal values are converted into the average power N, 27 of the noise signal in the device 25 for determining the average power.

In the second alternative embodiment, version 2, the average power S + N of the received signal values 14 is first calculated in the device 29. Subsequently, the average power of the signal part calculated in the device 24 is subtracted in the subtraction device 30. The average powers S and N or, respectively, 27 are conducted to the division device 28 that forms the ratio of the average powers of signal part S and noise part N, what is referred to as the signal-to-noise ratio (SNR) 22. This signal-to-noise ratio (SNR) indicates the quality of the transmission of digital information via the transmission channel. Since, differing from analog transmission signals, one does usually not speak of signal-to-noise ration or, respectively, signal-to-noise ratio SNR [sic] given digital transmission channels but generally utilizes the bit error rate or symbol error rate for evaluating the quality of a transmission system, a device 12 is inventively provided that converts the identified signal-to-noise ratio 22 into the generally standard symbol error rate (or bit error rate) 23. To that end, the identified SNR value 22 is converted into the desired symbol error rate 23 with a known mapping rule 24 in Fig. 5.

The mapping rule to be respectively employed is dependent on the encoding method and modulation method employed. A few known characteristics for converting the signal-to-noise ratio SNR into the probability for a symbol error P_M are shown in Fig. 6. Each characteristic thereby corresponds to a different encoding method. M thereby denotes the plurality of different possible signal values, QAM and PSK stand for different encoding methods; PSK denotes “phase shift keying” and QAM stands for quadrature amplitude demodulation.